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APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
09/720,767	12/29/2000	Kazunori Ozawa	P/126-197	5697

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DICKSTEIN SHAPIRO MORIN & OSHINSKY LLP
1177 AVENUE OF THE AMERICAS (6TH AVENUE)
41 ST FL.
NEW YORK, NY 10036-2714

EXAMINER	
HARPER, V PAUL	
ART UNIT	**PAPER NUMBER**
2654	

DATE MAILED: 11/01/2004

Please find below and/or attached an Office communication concerning this application or proceeding.

Office Action Summary	Application No. 09/720,767	Applicant(s) OZAWA, KAZUNORI	
	Examiner V. Paul Harper	Art Unit 2654	

-- *The MAILING DATE of this communication appears on the cover sheet with the correspondence address* --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If the period for reply specified above is less than thirty (30) days, a reply within the statutory minimum of thirty (30) days will be considered timely.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☐ Responsive to communication(s) filed on _____.

2a)☐ This action is **FINAL**. 2b)☒ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) 1-15 is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) 1-15 is/are rejected.

7)☒ Claim(s) 12 is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☐ The drawing(s) filed on _____ is/are: a)☐ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☒ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☒ All b)☐ Some * c)☐ None of:

1.☒ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)

2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)

3) ☒ Information Disclosure Statement(s) (PTO-1449 or PTO/SB/08) Paper No(s)/Mail Date 12/29/00,11/06/03, 3/29/01

4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____.

5) ☐ Notice of Informal Patent Application (PTO-152)

6) ☐ Other: _____.

DETAILED ACTION

Information Disclosure Statement

1. The Examiner has considered the references listed in the Information Disclosure Statements dated 12/29/2000, 3/29/2001, and 11/06/2003. Copies of these Information Disclosure Statements are attached to this office action.

Claim Objections

2. Claim 12 is objected to because it does not end in a period.

Appropriate correction is required.

Claim Rejections - 35 USC § 102

The following is a quotation of the appropriate paragraphs of 35 U.S.C. 102 that form the basis for the rejections under this section made in this Office action:

> A person shall be entitled to a patent unless –
>
> (b) the invention was patented or described in a printed publication in this or a foreign country or in public use or on sale in this country, more than one year prior to the date of application for patent in the United States.

3. Claims 1-6, 8 and 9 are rejected under 35 U.S.C. 102(b) as being anticipated by Ozawa (Japanese Patent Application Publication, 06-222797), hereinafter referred to as Ozawa.

Regarding **claim 1**, Ozawa discloses a voice encoding system that includes the following:

- a spectral parameter calculating unit supplied with a speech signal for calculating and quantizing spectral parameters (Drawing 2, item 200; ¶[0037]);
- an adaptive codebook unit for calculating a delay and a gain from a preceding quantized excitation signal by the use of an adaptive codebook, predicting the speech signal, and calculating a residue (Drawing 2, item 300; ¶[0057]); and
- an excitation quantizing unit for quantizing an excitation signal of said speech signal by the use of said spectral parameters to produce an output (Drawing 2, item 350; ¶[0062]);

said speech coder further comprising:

- a judging unit for extracting a feature from said speech signal to judge a mode (Drawing 2, item 245, ¶[0045]);
- a codebook for representing the excitation signal by a combination of a plurality of nonzero pulses and simultaneously quantizing amplitudes or polarities of said pulses in case where the output of said judging unit is a predetermined mode (Drawing 2, items 351_1, 351_N, also see equation 23);
- said excitation quantizing unit for searching combinations of code vectors stored in said codebook and a plurality of shift amounts for shifting pulse positions of said pulses and producing as an output a combination of the code vector and the shift amount, the produced combination minimizing distortion from an input speech (¶[0066], minimizes distortion); and
- a multiplexer unit for producing a combination of the output of said spectral parameter calculating unit, the output of said judging unit, the output of said

adaptive codebook unit, and the output of said excitation quantizing unit (Drawing 2, item 400).

Regarding **claim 2**, Ozawa discloses a voice encoding system that includes the following:

- a spectral parameter calculating unit supplied with a speech signal for calculating and quantizing spectral parameters (Drawing 2, item 200; ¶[0037]);
- an adaptive codebook unit for calculating a delay and a gain from a preceding quantized excitation signal by the use of an adaptive codebook, predicting a speech signal, and calculating a residue (Drawing 2, item 300; ¶[0057]); and
- an excitation quantizing unit for quantizing an excitation signal of said speech signal by the use of said spectral parameters to produce an output (Drawing 2, item 350; ¶[0062]);

said speech coder further comprising:

- a judging unit for extracting a feature from said speech signal to judge a mode (Drawing 2, item 245, ¶[0045]);
- a codebook for representing the excitation signal by a combination of a plurality of nonzero pulses and simultaneously quantizing amplitudes or polarities of said pulses in case where the output of said judging unit is a predetermined mode (Drawing 2, items 351_1, 351_N, also see equation 23);

- said excitation quantizing unit for generating pulse positions of said pulses in accordance with a predetermined rule and producing a code vector which minimizes distortion from the input speech (¶[0066], minimizes distortion); and
- a multiplexer unit for producing a combination of the output of said spectral parameter calculating unit, the output of said judging unit, the output of said adaptive codebook unit, and the output of said excitation quantizing unit (Drawing 2, item 400).

Regarding **claim 3**, Ozawa discloses a voice encoding system that includes the following:

- a spectral parameter calculating unit supplied with a speech signal for calculating and quantizing spectral parameters (Drawing 2, item 200; ¶[0037]);
- an adaptive codebook unit for calculating a delay and a gain from a preceding quantized excitation signal by the use of an adaptive codebook, predicting a speech signal, and calculating a residue (Drawing 2, item 300; ¶[0057]); and
- an excitation quantizing unit for quantizing an excitation signal of said speech signal by the use of said spectral parameters to produce an output (Drawing 2, item 350; ¶[0062]);

said speech coder comprising:

- a judging unit for extracting a feature from said speech signal to judge a mode (Drawing 2, item 245, ¶[0045]);

- a codebook for representing the excitation signal by a combination of a plurality of nonzero pulses and simultaneously quantizing amplitudes or polarities of said pulses in case where the output of said judging unit is a predetermined mode and a gain codebook for quantizing the gain (Drawing 2, items 351_1, 351_N, also see equation 23);
- said excitation quantizing unit for searching combinations of code vectors stored in said codebook, a plurality of shift amounts for shifting pulse positions of said pulses, and gain code vectors stored in said gain codebook, and producing as an output a combination of the code vector, the shift amount, and the gain code vector, the produced combination minimizing distortion from an input speech (¶[0062] – [0073], minimizes distortion); and
- a multiplexer unit for producing a combination of the output of said spectral parameter calculating unit, the output of said judging unit, the output of said adaptive codebook unit, and the output of said excitation quantizing unit (Drawing 2, item 400).

Regarding **claim 4**, Ozawa discloses a voice encoding system that includes the following:

- a spectral parameter calculating unit supplied with a speech signal for calculating and quantizing spectral parameters (Drawing 2, item 200; ¶[0037]);

- an adaptive codebook unit for calculating a delay and a gain from a preceding quantized excitation signal by the use of an adaptive codebook, predicting a speech signal, and calculating a residue (Drawing 2, item 300; ¶[0057]); and
- an excitation quantizing unit for quantizing an excitation signal of said speech signal by the use of said spectral parameters to produce an output (Drawing 2, item 350; ¶[0062]);

said speech coder comprising:

- a judging unit for extracting a feature from said speech signal to judge a mode (Drawing 2, item 245, ¶[0045]);
- a codebook for representing the excitation signal by a combination of a plurality of nonzero pulses and simultaneously quantizing amplitudes or polarities of said pulses in case where the output of said judging unit is a predetermined mode and a gain codebook for quantizing the gain (Drawing 2, items 351_1, 351_N, also see equation 23);
- said excitation quantizing unit for generating pulse positions of said pulses in accordance with a predetermined rule and producing a combination of the code vector and the gain code vector, the combination minimizing distortion from the input speech (¶[0062] – [0073], minimizes distortion); and
- a multiplexer unit for producing a combination of the output of said spectral parameter calculating unit, the output of said judging unit, the output of said adaptive codebook unit, and the output of said excitation quantizing unit (Drawing 2, item 400).

Regarding **claim 5**, Ozawa discloses a voice encoding system that includes the following:

- spectral parameter calculating means supplied with a speech signal for calculating and quantizing spectral parameters (Drawing 2, item 200; ¶[0037]);
- adaptive codebook means for calculating a delay and a gain from a preceding quantized excitation signal by the use of an adaptive codebook, predicting a speech signal, and calculating a residue (Drawing 2, item 300; ¶[0057]);
- mode judging means for extracting a feature quantity from said speech signal and carrying out mode judgment as to the utterance or the silence and so on (Drawing 2, item 245, ¶[0045]);
- excitation quantizing means for quantizing an excitation signal of said speech signal by the use of said spectral parameters to produce an output, said excitation quantizing means searching, in case of a predetermined mode, combinations of code vectors stored in a codebook for simultaneously quantizing amplitudes or polarities of a plurality of pulses and a plurality of shift amounts for temporally shifting predetermined positions of the pulses and selecting a combination of the index of the code vector and the shift amount, the selected combination minimizing distortion from an input speech (Drawing 2, item 350; ¶[0062] – [0073], minimizes distortion);
- gain quantizing means for quantizing the gain by the use of a gain codebook (Drawing 2, item 355; ¶[0067]); and

- multiplexer means for producing a combination of the outputs of said spectral parameter calculating means, said adaptive codebook means, said excitation quantizing means, and said gain quantizing means (Drawing 2, item 400).

Regarding **claim 6**, Ozawa teaches everything claimed, as applied above (see claim 5); in addition, Ozawa teaches "said excitation quantizing means uses, as the pulse positions, positions generated in accordance with a predetermined rule in case where judgment by said mode judging means indicates a predetermined mode" (¶[0057] pitch parameter determines pulse positions).

Regarding **claim 8**, Ozawa teaches everything claimed, as applied above (see claim 5); in addition, Ozawa teaches the following:

- said excitation quantizing means selects, from all combinations of every code vectors in said codebook and every shift amounts for the pulse positions, a plurality of combinations in the order of minimizing a predefined distortion and delivers the combinations to said gain quantizing means, in case where judgment in said mode judging means indicates a predetermined mode (Drawing 2, item 350; ¶[0062]; formula 23, also note input from item 245, mode classifier);
- said gain quantizing means quantizing the gain by the use of said gain codebook for each of a plurality of sets of the outputs supplied from said excitation quantizing means and selecting a combination of the shift amount, the excitation

code vector, and the gain code vector, the combination minimizing the predetermined distortion (Drawing 2, item 355; ¶[0067]-[0070]).

Regarding **claim 9**, Ozawa teaches everything claimed, as applied above (see claim 5); in addition, Ozawa teaches "said mode judging means uses a pitch prediction gain as the feature quantity of said speech signal, compares the value of the pitch prediction gain calculated for each subframe and a predetermined threshold value, and judges the utterance and the silence when the pitch prediction gain is greater and smaller than said threshold value, respectively" (¶[0027]-[0031]).

Claim Rejections - 35 USC § 103

The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

4. Claim 7 is rejected under 35 U.S.C. 103(a) as being unpatentable over Ozawa in view of Paksoy et al. (U.S. Patent 6,148,282), hereinafter referred to as Paksoy.

Regarding **claim 7,** Ozawa teaches everything claimed, as applied above (see claim 5). In addition, Ozawa teaches the mode classification of the signal into two or more modes (¶[0006]), but Ozawa does not specifically teach the use of a "random

number generating means for generating a predetermined number of pulse positions, said random number generating means delivering said positions thus generated to said excitation quantizing means in case where judgment by said mode judging means indicates a predetermined mode." However, the examiner contends that this concept was well known in the art, as taught by Paksoy.

In the same field of endeavor, Paksoy discloses a multimodal code-excited linear prediction (CELP) coder that includes a mode characterized by random excitation (col. 3, lines 47-64).

Therefore, it would have been obvious to one having ordinary skill in the art at the time the invention was made to modify Ozawa by specifically providing the technique, as taught by Paksoy, because it is well known in the art at the time of invention for the purpose of efficiently representing the signal.

5. Claims 10-15 are rejected under 35 U.S.C. 103(a) as being unpatentable over Ozawa in view of Atal (U.S. Patent 4,220,819), hereinafter referred to as Atal.

Regarding **claim 10**, Ozawa teaches everything claimed, as applied above (see claim 5). In addition, Ozawa teaches the mode classification of the signal into two or more modes (¶[0006]), but Ozawa does not specifically teach the "said predetermined mode is silence." However, the examiner contends that this concept was well known in the art, as taught by Atal.

In the same field of endeavor, Atal discloses a residual excited predictive speech coding system, which includes the classification of the signal as silence (col. 5, lines 50-65; col. 23, code for **VOICED-UNVOICED-SILENCE DECISION**)

Therefore, it would have been obvious to one having ordinary skill in the art at the time the invention was made to modify Ozawa by specifically providing the technique, as taught by Atal, because it is well known in the art at the time of invention for the purpose of efficiently representing the signal.

Regarding **claim 11**, this claim has coder limitations similar to those in claim 1 and is rejected for the same reasons. Furthermore, Ozawa does not specifically teach the following decoder limitations:

a) demultiplexer means supplied with a coded output of said speech coder for demultiplexing the coded output into codes representative of spectral parameters, delays of said adaptive codebook, adaptive code vectors, excitation gains, amplitudes or polarity code vectors as excitation information, and pulse positions and delivering these codes;
b) mode judging means for judging a mode by the use of a preceding quantized gain in an adaptive codebook;
c) excitation signal restoring means for generating, in case where the output of said mode judging means is a predetermined mode, pulse positions in accordance with a predefined rule, generating amplitudes or polarities of said pulses from the code vectors, and restoring an excitation signal; and

d) a synthesis filter unit for passing said excitation signal to reproduce a speech signal.

However, the examiner contends that a particular set of encoding operations necessarily requires a corresponding set of decoding operations, and that this concept was well known in the art, as taught by Atal.

In the same field of endeavor, Atal discloses a residual excited predictive speech coding system with an encoder (Fig. 1) and a corresponding (and necessary) decoder (Fig. 2). Furthermore, at the unit level, Atal teaches the use of a spectral signal encoder (unit) within the encoder (item 126) where the decoder has a corresponding spectral signal decoder (unit) (item 203). This implies that limitations a)-c), above, are obvious since each of these decoder units corresponds to a claimed coder unit and without such a correspondence the encoder/decoder system would not have utility. Additionally, Atal teaches the use of an LPC Synthesizer (Fig. 2, item 230) which corresponds to limitation d), above.

Therefore, it would have been obvious to one having ordinary skill in the art at the time the invention was made to modify Ozawa by specifically providing the corresponding decoder functionality, and the synthesis technique, as taught by Atal, because it is well known in the art at the time of invention as the conventional way to implement an encoder/decoder system.

Regarding **claim 12**, this claim has coder limitations similar to those in claim 2 and is rejected for the same reasons. Furthermore, Ozawa does not specifically teach the following decoder limitations:

a) demultiplexer means supplied with a coded output of said speech coder for demultiplexing the coded output into codes representative of spectral parameters, delays of said adaptive codebook, adaptive code vectors, excitation gains, amplitudes or polarity code vectors as excitation information, and pulse positions and outputting these codes;

b) mode judging means for judging a mode by the use of a preceding quantized gain in an adaptive codebook;

c) excitation signal restoring means for generating, in case where the output of said mode judging means is the predetermined mode, the pulse positions in accordance with a predefined rule, generating amplitudes or polarities of said pulses from code vectors, and restoring an excitation signal; and

d) a synthesis filter unit for passing said excitation signal to reproduce a speech signal

However, the examiner contends that a particular set of encoding operations necessarily requires a corresponding set of decoding operations, and that this concept was well known in the art, as taught by Atal.

In the same field of endeavor, Atal discloses a residual excited predictive speech coding system with an encoder (Fig. 1) and a corresponding (and necessary) decoder (Fig. 2). Furthermore, at the unit level, Atal teaches the use of a spectral signal encoder

(unit) within the encoder (item 126) where the decoder has a corresponding spectral signal decoder (unit) (item 203). This implies that limitations a)-c), above, are obvious since each of these decoder units corresponds to a claimed coder unit and without such a correspondence the encoder/decoder system would not have utility. Additionally, Atal teaches the use of an LPC Synthesizer (Fig. 2, item 230) which corresponds to limitation d), above.

Therefore, it would have been obvious to one having ordinary skill in the art at the time the invention was made to modify Ozawa by specifically providing the corresponding decoder functionality, and the synthesis technique, as taught by Atal, because it is well known in the art at the time of invention as the conventional way to implement an encoder/decoder system.

Regarding **claim 13**, this claim has coder limitations similar to those in claim 3 and is rejected for the same reasons. Furthermore, Ozawa does not specifically teach the following decoder limitations:

a) demultiplexer means supplied with a coded output of said speech coder for demultiplexing the coded output into codes representative of spectral parameters, delays of said adaptive codebook, adaptive code vectors, excitation gains, amplitudes or polarity code vectors as excitation information, and pulse positions and delivering these codes;
b) mode judging means for judging a mode by the use of a preceding quantized gain in an adaptive codebook;

c) excitation signal restoring means for generating in case where the output of said mode judging means is the predetermined mode, pulse positions in accordance with a predefined rule, generating amplitudes or polarities of said pulses from code vectors, and restoring an excitation signal; and

d) a synthesis filter unit for passing said excitation signal to reproduce a speech signal.

However, the examiner contends that a particular set of encoding operations necessarily requires a corresponding set of decoding operations, and that this concept was well known in the art, as taught by Atal.

In the same field of endeavor, Atal discloses a residual excited predictive speech coding system with an encoder (Fig. 1) and a corresponding (and necessary) decoder (Fig. 2). Furthermore, at the unit level, Atal teaches the use of a spectral signal encoder (unit) within the encoder (item 126) where the decoder has a corresponding spectral signal decoder (unit) (item 203). This implies that limitations a)-c), above, are obvious since each of these decoder units corresponds to a claimed coder unit and without such a correspondence the encoder/decoder system would not have utility. Additionally, Atal teaches the use of an LPC Synthesizer (Fig. 2, item 230) which corresponds to limitation d), above.

Therefore, it would have been obvious to one having ordinary skill in the art at the time the invention was made to modify Ozawa by specifically providing the corresponding decoder functionality, and the synthesis technique, as taught by Atal,

because it is well known in the art at the time of invention as the conventional way to implement an encoder/decoder system.

Regarding **claim 14**, this claim has coder limitations similar to those in claim 4 and is rejected for the same reasons. Furthermore, Ozawa does not specifically teach the following decoder limitations:

a) demultiplexer means supplied with a coded output of said speech coder for demultiplexing the coded output into codes representative of spectral parameters, delays of said adaptive codebook, adaptive code vectors, excitation gains, amplitudes or polarity code vectors as excitation information, and pulse positions and delivering these codes;
b) mode judging means for judging a mode by the use of a preceding quantized gain in an adaptive codebook;
c) excitation signal restoring means for generating, in case where the output of said mode judging means is the predetermined mode, pulse positions in accordance with a predefined rule, generating amplitudes or polarities of said pulses from code vectors, and restoring an excitation signal; and
d) a synthesis filter unit for passing said excitation signal to reproduce a speech signal.

However, the examiner contends that a particular set of encoding operations necessarily requires a corresponding set of decoding operations, and that this concept was well known in the art, as taught by Atal.

In the same field of endeavor, Atal discloses a residual excited predictive speech coding system with an encoder (Fig. 1) and a corresponding (and necessary) decoder (Fig. 2). Furthermore, at the unit level, Atal teaches the use of a spectral signal encoder (unit) within the encoder (item 126) where the decoder has a corresponding spectral signal decoder (unit) (item 203). This implies that limitations a)-c), above, are obvious since each of these decoder units corresponds to a claimed coder unit and without such a correspondence the encoder/decoder system would not have utility. Additionally, Atal teaches the use of an LPC Synthesizer (Fig. 2, item 230) which corresponds to limitation d), above.

Therefore, it would have been obvious to one having ordinary skill in the art at the time the invention was made to modify Ozawa by specifically providing the corresponding decoder functionality, and the synthesis technique, as taught by Atal, because it is well known in the art at the time of invention as the conventional way to implement an encoder/decoder system.

Regarding **claim 15**, this claim has coder limitations similar to those in claim 5 and is rejected for the same reasons. Furthermore, Ozawa does not specifically teach the following decoder limitations:

a) demultiplexer means supplied with a coded output of said speech coder for demultiplexing the coded output into codes representative of spectral parameters, delays of said adaptive codebook, adaptive code vectors, excitation

gains, amplitudes or polarity code vectors as excitation information, and pulse positions and delivering these codes;

b) mode judging means for judging a mode by the use of a preceding quantized gain in an adaptive codebook;

c) excitation signal restoring means for generating, in case where the output of said mode judging means is the predetermined mode, pulse positions in accordance with a predefined rule, generating amplitudes or polarities of said pulses from code vectors, and restoring an excitation signal; and

d) a synthesis filter unit for passing said excitation signal to reproduce a speech signal.

However, the examiner contends that a particular set of encoding operations necessarily requires a corresponding set of decoding operations, and that this concept was well known in the art, as taught by Atal.

In the same field of endeavor, Atal discloses a residual excited predictive speech coding system with an encoder (Fig. 1) and a corresponding (and necessary) decoder (Fig. 2). Furthermore, at the unit level, Atal teaches the use of a spectral signal encoder (unit) within the encoder (item 126) where the decoder has a corresponding spectral signal decoder (unit) (item 203). This implies that limitations a)-c), above, are obvious since each of these decoder units corresponds to a claimed coder unit and without such a correspondence the encoder/decoder system would not have utility. Additionally, Atal teaches the use of an LPC Synthesizer (Fig. 2, item 230) which corresponds to limitation d), above.

Therefore, it would have been obvious to one having ordinary skill in the art at the time the invention was made to modify Ozawa by specifically providing the corresponding decoder functionality, and the synthesis technique, as taught by Atal, because it is well known in the art at the time of invention as the conventional way to implement an encoder/decoder system.

Conclusion

Any inquiry concerning this communication or earlier communications from the examiner should be directed to V. Paul Harper whose telephone number is 703 305-4197. The examiner can normally be reached on M-F.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Richemond Dorvil can be reached on 703 305-9645. The fax phone number for the organization where this application or proceeding is assigned is 703-872-9306.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free).

V. Paul Harper

VPH/vph

Vijay Chawan 10/28/04

VIJAY CHAWAN
PRIMARY EXAMINER